Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP IP Telephony Using Avaya 4600 Series IP Telephones, Avaya one-X™ Desktop Edition, and Cisco SIP Proxy Server – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to configure Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition with the Cisco SIP Proxy Server. The Cisco SIP Proxy Server supports the standard SIP features and some supplementary SIP features. The Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition also support the standard SIP features and some supplementary SIP features. The Avaya one-X™ Desktop Edition also supports presence and instant messaging. Testing was conducted at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the configuration steps for using Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition[1] with the Cisco SIP Proxy Server. Only those configuration steps pertinent to interoperability of the Cisco and Avaya equipment are covered. General administration information can be found in the product documentation as well as the specific references listed in **Section 9**.

The configuration used is shown in **Figure 1**. All components are physically connected to a single Avaya C363T-PWR Converged Stackable Switch and are administered as a single subnet. A PC provides HTTP/HTTPS, DHCP, and TFTP server support. The Avaya one-X™ Desktop Edition clients run on separate PCs running Microsoft Windows XP Professional operating system.

The main difference among the four Avaya 4600 Series IP Telephones (4602SW, 4610SW, 4620SW, & 4621SW), for SIP functionality, is the number of line appearances supported by each phone (two line appearances for the 4602SW and five for the 4610SW, 4620SW, and 4621SW). The other major differences are that the 4610SW, 4620SW, and 4621SW telephones provide call log and speed dial applications and a variety of options to view and/or modify the current settings for the phone. The 4602SW does not support these applications and options. Avaya one-X™ Desktop Edition supports up to ten line appearances and provides a Windows-based graphical user interface for access to call log, speed dials, and settings.

The configuration steps described in these Application Notes apply to all four models of the Avaya 4600 Series IP Telephones and to Avaya one-X™ Desktop Edition. **Table 1** profiles the network management capabilities of the phones.

[1] "Avaya one-X™ Desktop Edition" is the new brand for "Avaya SIP Softphone". Therefore, that name is used in these Application Notes to document the product. However, the version of the product that was used for testing still has internal references to "Avaya SIP Softphone" or "SIP Softphone".

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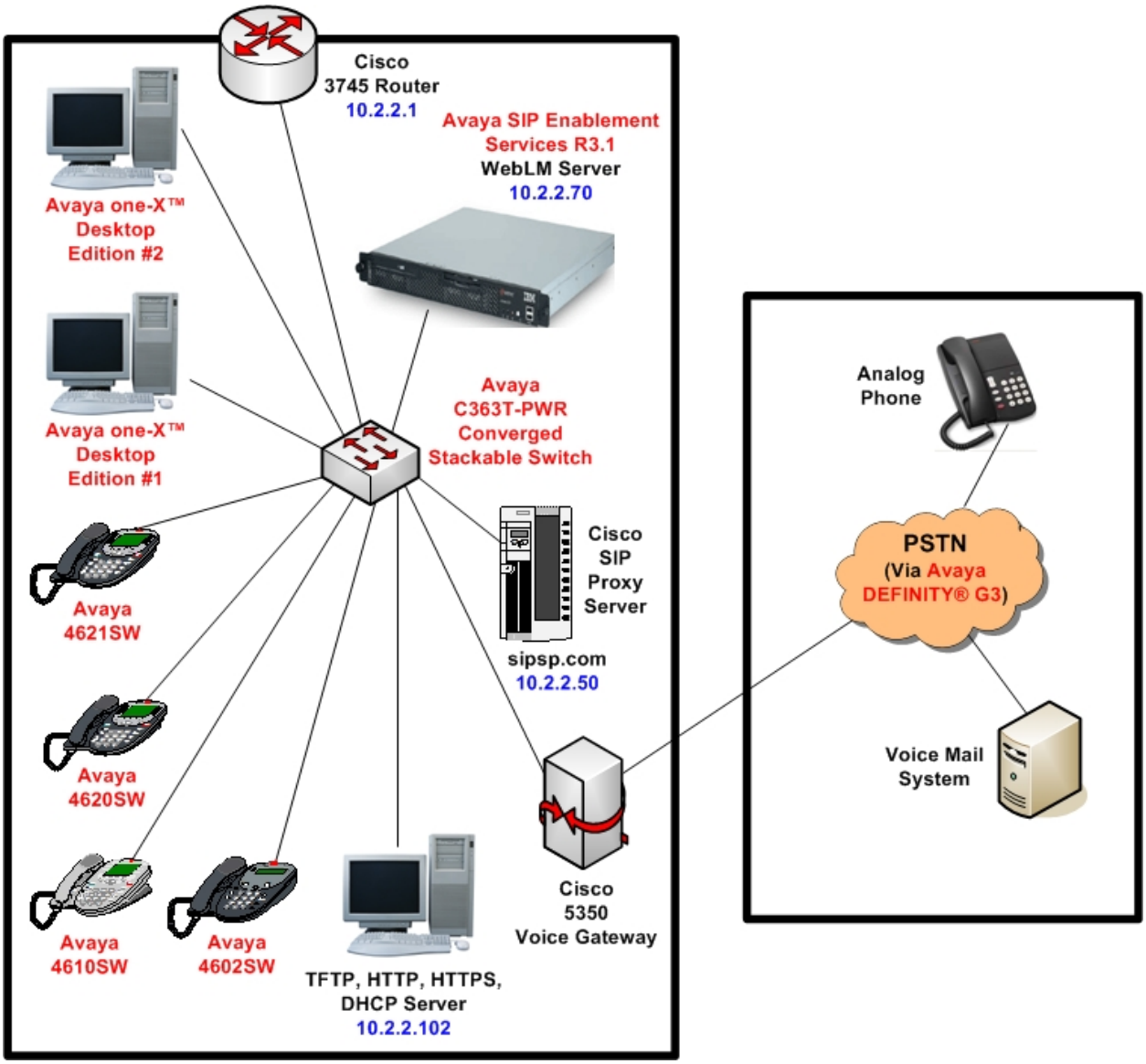

Figure 1: Network Configuration Diagram

	Avaya 4600 Series IP Telephones	**Avaya one-X™ Desktop Edition**
Administration mechanisms	Configuration files	Manual, Configuration files
File transfer server	TFTP, HTTP, HTTPS	DHCP, HTTP

Table 1: Network Management Capabilities of Avaya 4600 Series IP Telephones & Avaya one-X™ Desktop Edition

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2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya C363T-PWR Converged Stackable Switch	4.5.14
Avaya 4602SW, 4610SW, 4620SW, 4621SW IP Telephones	2.2.2
Avaya one-X™ Desktop Edition	2.1 (Load 43)
Avaya SIP Enablement Services (SES) for WebLM	3.1
Cisco SIP Proxy Server (CSPS)	2.2.1.1
Cisco 3745 Router	12.2(8r)T2
Cisco 5350 Voice Gateway	IOS 12.4(7)
Windows PCs for Avaya one-X™ Desktop Edition	Microsoft Windows XP 2002 SP2
Windows Server PC for DHCP, HTTP/HTTPS, TFTP	Microsoft Windows 2000 Server SP4
Microsoft Internet Explorer web browser	6.0.2900.2180.xpsp_sp2_gdr.050301-1519
Microsoft Internet Information Services (IIS)	5.0
PumpKIN TFTP Server	2.00

Table 2: Equipment and Version

3. Supported SIP Features

In addition to standard calling capabilities (i.e., make a call, answer a call, drop a call), the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features, often referred to as the SIPPING-19 [4]. This reference provides a useful framework to describe product capabilities and compare features supported by various equipment vendors. **Table 3** provides a summary of SIP features supported on the Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition when connected to the Cisco SIP Proxy Server.

Section 5 and **Section 6** of these Application Notes describes the steps for configuring the Avaya clients and Cisco server to support the SIP features.

NOTE: The features that were tested for these Applications Notes are listed as "**Yes**" in the "**Supported**" column in **Table 3**. "**Yes**" denotes that the feature is implemented by the Cisco server and by the Avaya phone device. "**N/A**" denotes that this feature is implemented by the Cisco server but is not implemented by the Avaya phone device. "**No**" denotes that this feature is not implemented by the Cisco server.

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#	Feature	Supported		Comments
		Avaya 4600 Series IP Telephones	Avaya one-X™ Desktop Edition	
Supplementary Features				
1.	**Call Hold**	**Yes**	**Yes**	
2.	**Consultation Hold**	**Yes**	**Yes**	
3.	Music On Hold	No	No	Cisco SIP proxy server does not include voice mail.
4.	**Unattended Transfer**	**Yes**	**Yes**	
5.	**Attended Transfer**	**Yes**	**Yes**	
6.	Transfer – Instant Messaging	No	No	Not applicable for Avaya 4600 Series IP Telephones; not supported by Avaya one-X™ Desktop Edition
7.	**Call Forward Unconditional – *via the server***	**Yes**	**Yes**	
	Call Forward Unconditional – *via the phone*	**Yes**	N/A	Not supported by the Avaya 4602SW or Avaya one-X™ Desktop Edition
8.	**Call Forward Busy – *via the server***	**Yes**	**Yes**	
	Call Forward Busy – *via the phone*	**Yes**	N/A	Not supported by the Avaya 4602SW or Avaya one-X™ Desktop Edition
9.	**Call Forward No Answer – *via the server***	**Yes**	**Yes**	
	Call Forward No Answer – *via the phone*	**Yes**	N/A	Not supported by the Avaya 4602SW or Avaya one-X™ Desktop Edition
10.	**3-way conference – 3rd party added by user**	**Yes**	**Yes**	
11.	**3-way conference – 3rd party calls and is joined by user**	**Yes**	**Yes**	
12.	Single Line Extension (forking)	No	No	Not supported by Cisco SIP proxy server[2]
13.	Find-me	No	No	
14.	Incoming Call Screening	No	No	
15.	Outgoing Call Screening	No	No	
16.	Call Park	No	No	
17.	Call Pick-up	No	No	
18.	Automatic Redial – *via the server*	No	No	
	Automatic Redial – *via the phone*	**Yes**	**Yes**	
19.	Click to Dial – *via the server*	No	No	
	Click to Dial – *via the phone*	N/A	**Yes**	Not supported by the Avaya 4600 Series IP Telephones
Other Features				
20.	Message Waiting Indicator (MWI)	No	No	Cisco SIP proxy server does not include voice mail.
21.	**Presence Tracking**	N/A	**Yes**	Not supported by the Avaya 4600 Series IP Telephones
22.	**Instant Messaging**	N/A	**Yes**	Not supported by the Avaya 4600 Series IP Telephones
23.	**DTMF support**	**Yes**	**Yes**	

Table 3: SIP Feature Support with Cisco SIP Proxy Server

[2] The Cisco proxy server supports serial and parallel forking but not the full feature set needed to support Single Line Extension.

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3.1.1. Call Forwarding

In addition to the call forwarding features provided by the Cisco server, the Avaya 4600 Series IP Telephones, except for the 4602SW, support local call forwarding. (Avaya one-X™ Desktop Edition does not support local call forwarding.) These features can be enabled via the **`46xxsettings.txt`** configuration file (see **Section 5**).

3.1.2. Message Waiting Indicator (MWI)

This feature was not tested as there is no voice mail system in the configuration listed in **Figure 1**.

3.1.3. Click to Dial

While there is a click to dial application supported with Avaya one-X™ Desktop Edition, this implementation uses a direct link from Microsoft Internet Explorer (using an Internet Explorer Browser Helper Object) to Avaya one-X™ Desktop Edition (see **Section 6.4** for the configuration of this feature). This implementation is not the same as the Click to Dial feature defined for SIP.

3.1.4. DTMF

For these Application Notes, SIP DTMF signaling (defined in RFC 2833) is tested via interaction with an external voice mail system. The ability for a voice mail subscriber to log into the system and to invoke other features of the system is dependent on the reception of DTMF. Calls were made via the Cisco voice gateway to the PSTN and out to an external voice mail system, as shown in **Figure 1**.

3.1.5. Presence Tracking and Instant Messaging

SIP Presence and Instant Messaging (IETF RFC 3856, RFC 2779, and SIMPLE [SIP for Instant Messaging and Presence Leveraging Extensions]) are supported by Avaya one-X™ Desktop Edition but not by the Avaya 4600 Series IP Telephones.

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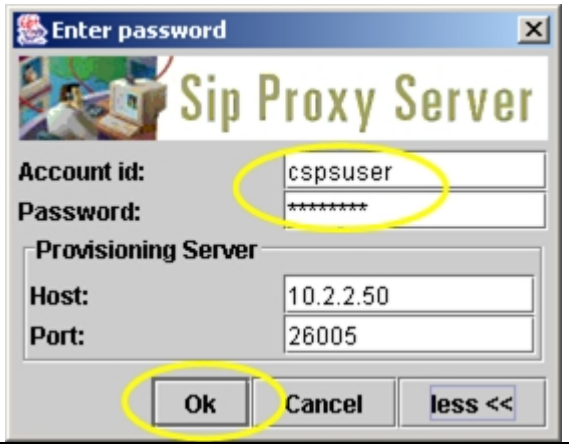
4. Configure the Cisco SIP Proxy Server

The following steps describe configuration of the Cisco SIP Proxy Server for use with the Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition. Other standard administration functions are covered in Reference [5]. The installation of the server and the GUI-based provisioning system is covered in Reference [6].

4.1. Administer the Cisco Proxy Server

Steps	Description
1.	The Cisco SIP proxy server is configured using a GUI-based provisioning system that can be installed on Microsoft Windows, Red Hat Linux, or Sun Solaris. The Microsoft Windows version was used for these Application Notes. However, as this is a Java application, the functionality is the same on all operating systems. Start the Cisco program by double clicking on the shortcut placed on the Windows desktop. Log into the system using the default account ***cspsuser*** and enter the appropriate password for this account. (The Host address and Port fields are only visible upon clicking on the ***more***>>> button. These fields contain the information that was entered on installation of the provisioning system.) Click the **Ok** button to continue. 

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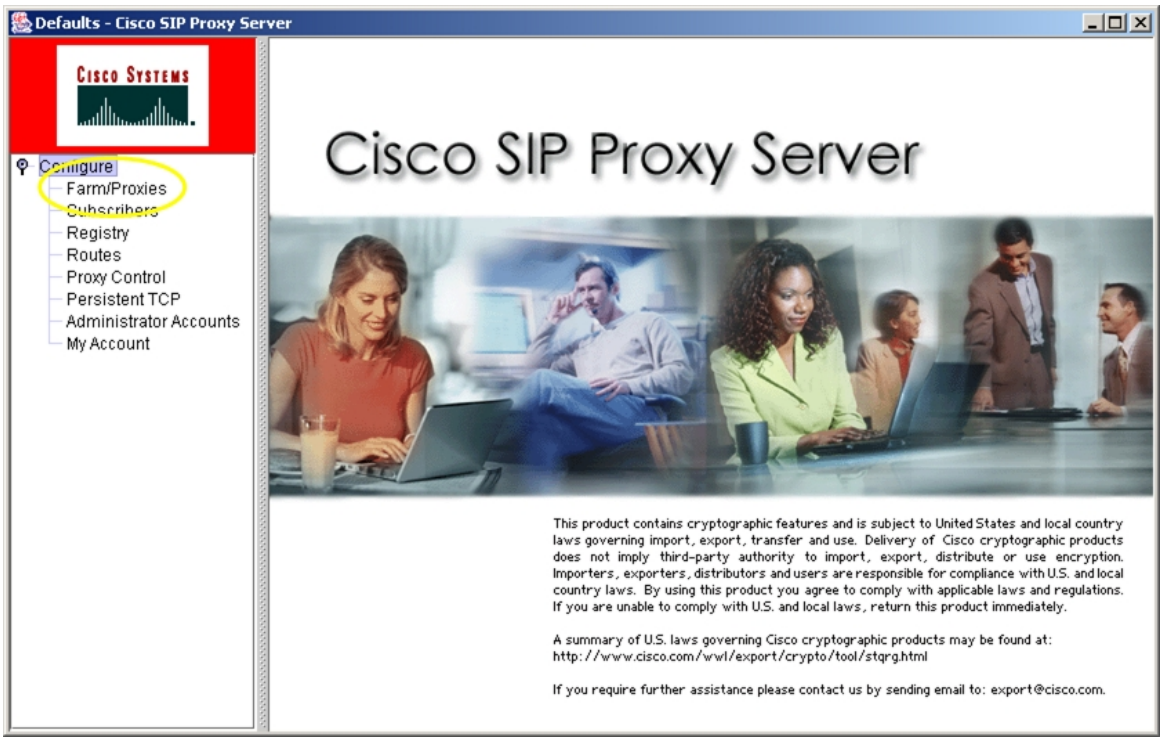
Steps	Description
2.	The *Cisco SIP Proxy Server* provisioning interface will be displayed. Click on the **Farm/Proxies** link on the left side of the page. 

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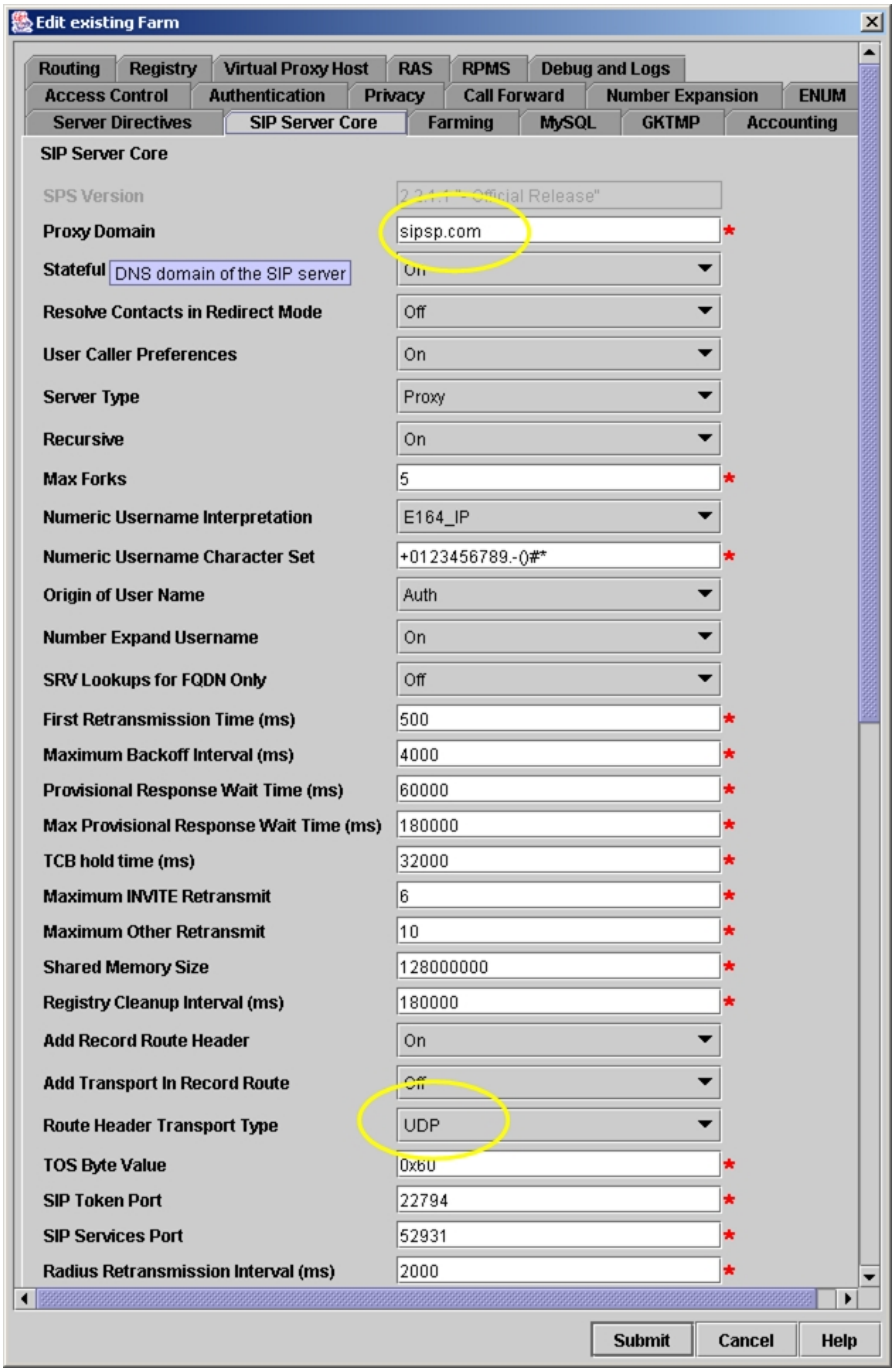
Steps	Description
3.	The *Edit existing Farm* page will be displayed. Click on the **SIP Server Core** tab at the top of the page. Fill in the required fields (indicated by *). In the screen below, the **Proxy Domain** is set to *sipsp.com*. The values indicated below and/or the system defaults can be used for the remaining fields. Make certain that the **Route Header Transport Type** is set to *UDP* as this is what is supported by the Avaya 4600 Series IP Telephones. Click on the **Authentication** tab at the top of the page. 

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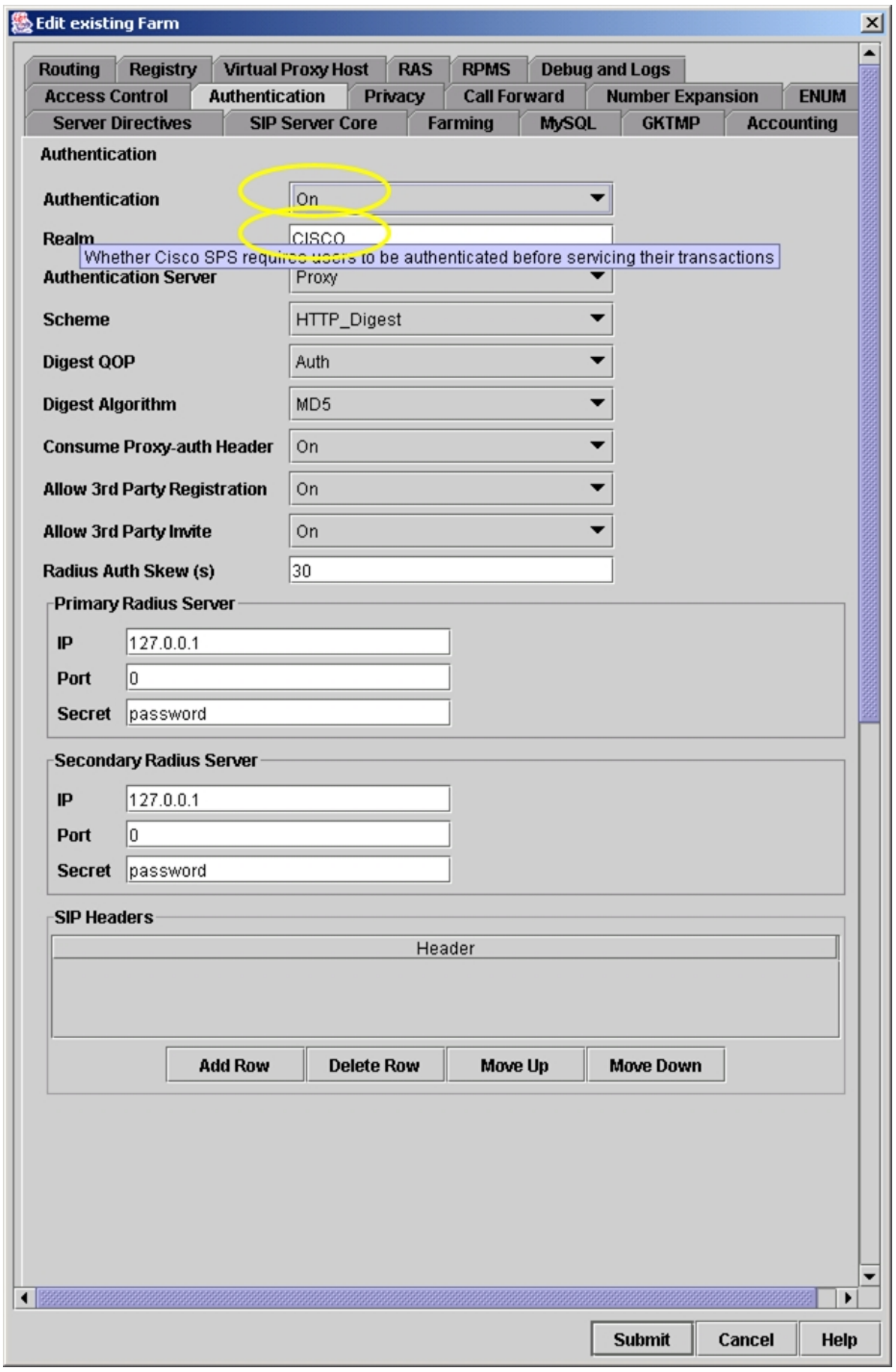
Steps	Description
4.	The *Authentication* tab in the *Edit existing Farm* page is displayed. Select *On* from the pull down list for the **Authentication** field. Enter *CISCO* in the **Realm** field. The values indicated below and/or the system defaults can be used for the remaining fields. Click on the **Call Forward** tab at the top of the page. 

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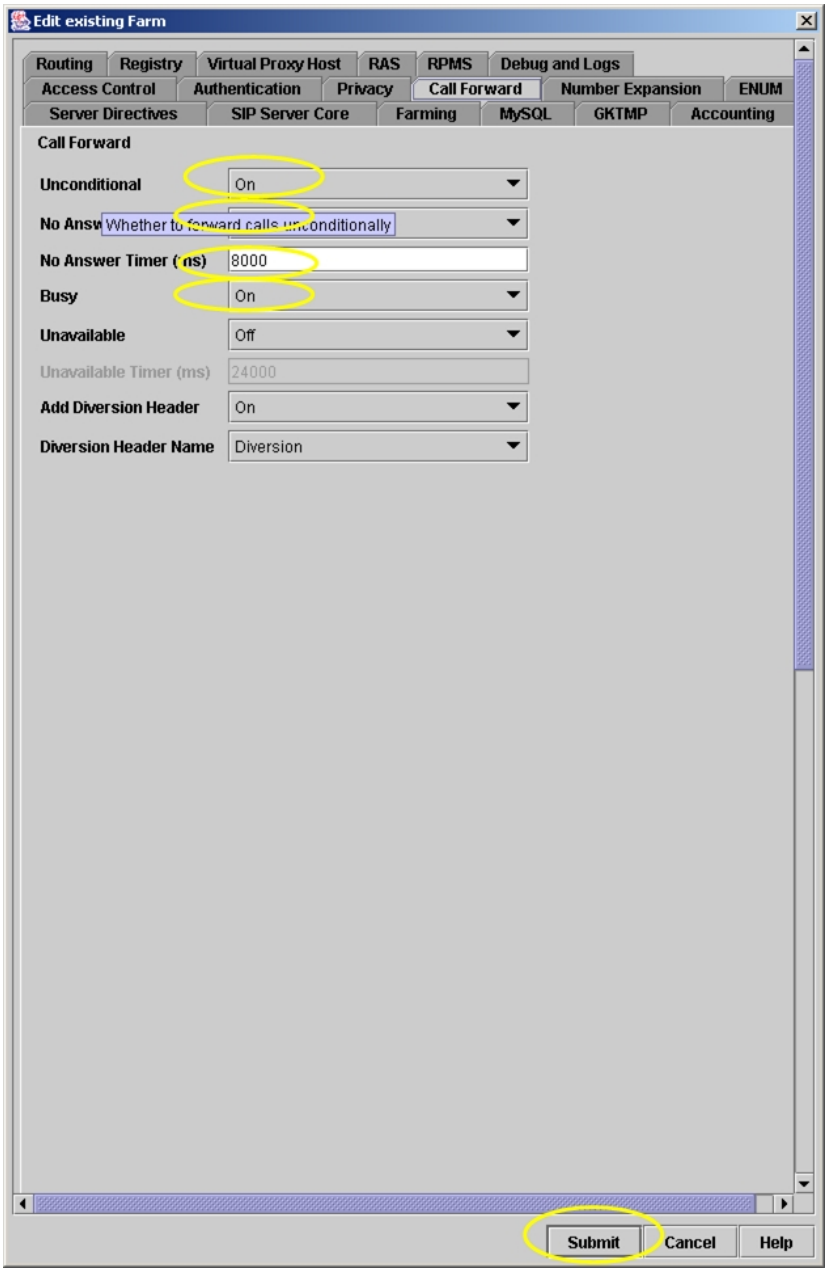
Steps	Description
5.	The *Call Forward* tab in the *Edit existing Farm* page is displayed. Select *On* from the pull down list for the **Unconditional**, the **No Answer**, and the **Busy** fields. In addition, change the value in the **No Answer Timer (ms)** field to decrease or increase time to wait before forwarding takes place when the phone is not answered. The values indicated below and/or the system defaults can be used for the remaining fields. Click the **Submit** button to save the changes. 

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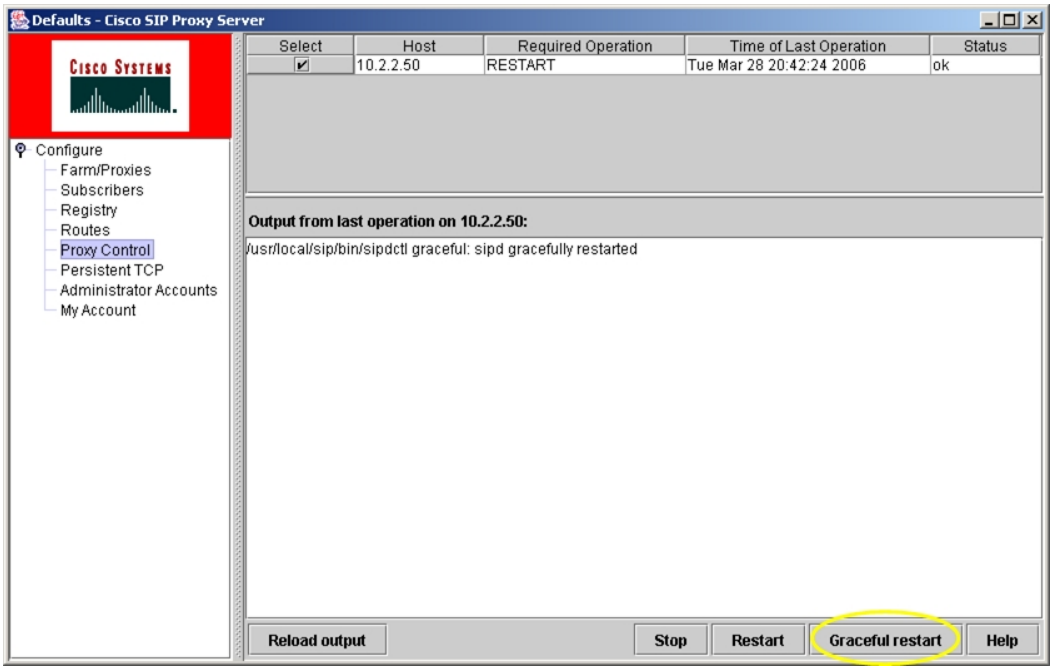
Steps	Description
6.	The *Data written* dialogue is displayed. The configuration changes need to be saved to the database and the server must be restarted with the new values. Click on the **Go to Proxy Control screen** button to go to restart the server.
7.	The *Cisco SIP Proxy Server* page is displayed with the Proxy Control shown. Click on the **Graceful restart** button to restart the server. 

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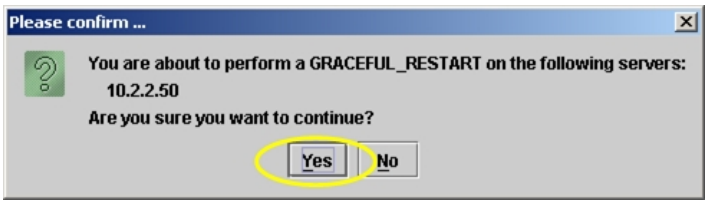
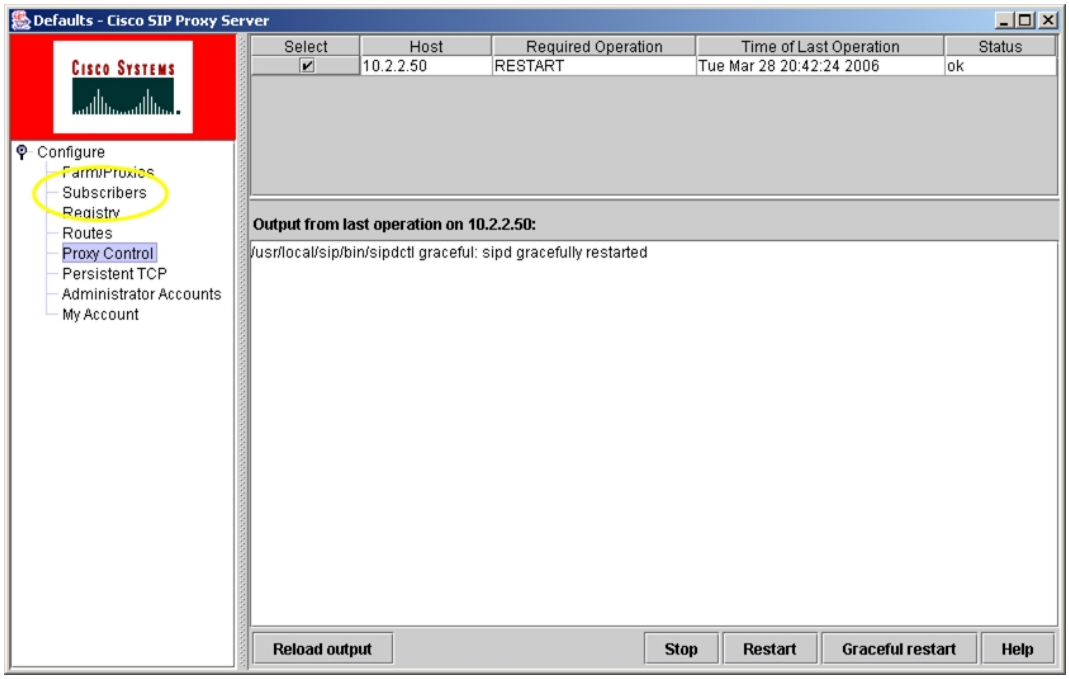
Steps	Description
8.	The *Please confirm …* dialogue is displayed. Click on the **Yes** button to confirm that the server should be restarted. 
9.	The *Cisco SIP Proxy Server* page is displayed with the Proxy Control shown. The message in this page will indicate that the server was gracefully restarted. The basic server administration is complete. Continue to the next section to administer users, otherwise know as subscribers on the Cisco system. Click on **Subscribers**. 

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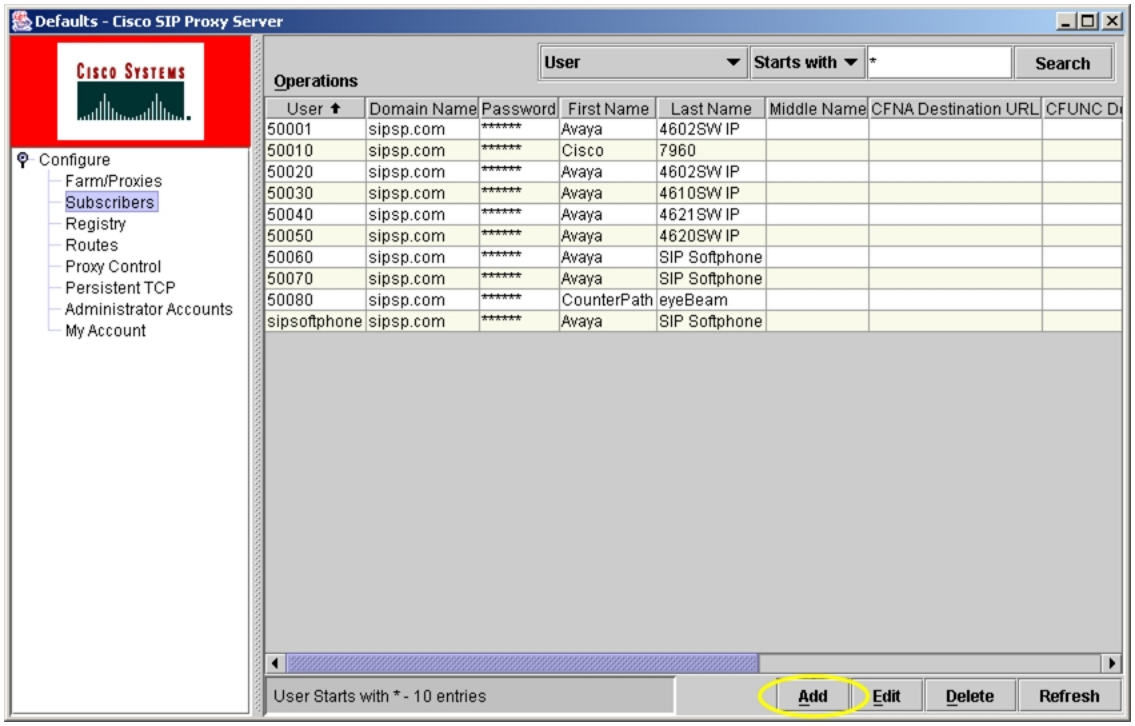
4.2. Administer Users (Subscribers) on the Cisco Proxy Server

The following steps describe administration of the Cisco SIP proxy server for use with Avaya SIP clients.

Steps	Description
1.	The *Cisco SIP Proxy Server* provisioning interface is displayed. Click on **Add** to administer new subscribers. 

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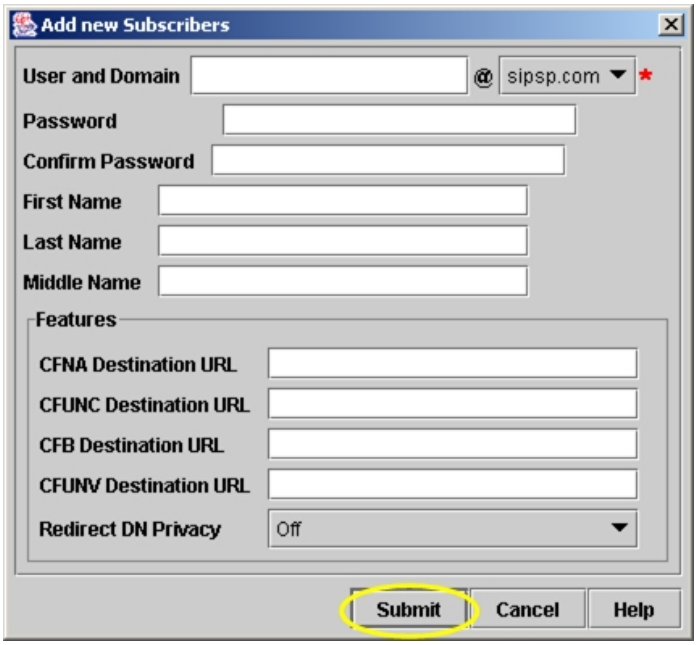
Steps	Description
2.	The *Add new Subscribers* page is displayed. Fill in the following fields: ▪ **User and Domain**: Enter an numeric extension, e.g., "50000" ▪ **Password**: Enter a password, e.g., "123456" ▪ **Confirm Password**: Re-enter the password ▪ **First Name**: Enter a first name, e.g., "Avaya" ▪ **Last Name**: Enter a last name, e.g., "4602SW" Optionally, configure the call forwarding feature for this subscriber. "Call Forwarding No Answer" (**CFNA Destination URL**) and "Call Forwarding on Busy" (**CFB Destination URL**) can be configured at the same time. Configure "Call Forwarding Unconditional" (**CFUNC Destination URL**) only when ready to test that feature. Remove the **CFUNC** configuration to test the other call forwarding features and/or to receive calls at this station. The call forwarding fields must be filled with a URL. For example, to forward to extension 50010, enter *sip:50010@sipsp.com*. Click on **Submit** to save the changes to the database. 

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5. Configure the Avaya 4600 IP Telephones

The SIP software should be installed in the Avaya 4600 Series IP Telephones using the procedures described in Reference [1]. The SIP specific software can be downloaded from the Avaya Support Center site (http://www.avaya.com/support). Download the **`46xxSIP*.zip`** file (where **`*`** is the date) and install the files per the instructions in Reference [1]. Any Avaya 4600 Series IP Telephone that has the H.323 software loaded must be optioned to install the SIP software by modifying the **`SIG`** value to **`SIP`** (press **<MUTE>-S-I-G** on the telephone keypad) and restarting the telephone.

The installation of the telephone software and the configuration of the telephone require the use of a DHCP server, a TFTP server, and an HTTP/HTTPS server. These Application Notes will not provide the details of the administration of Avaya 4600 Series IP Telephones (see Reference [1]). These Application Notes will provide details of the specific SIP related configuration items.

Steps	Description
1.	Locate the **`46xxsettings.txt`** file to administer the SIP specific settings for the Avaya 4600 Series IP Telephones. This file is stored on the PC that provides TFTP and HTTP/HTTPS services to the telephones. The location of this file depends on the software that is used to provide these services. For these Application Notes, the file is stored on the **`Inetpub\wwwrooot`** folder on the Windows server PC. The **`46xxsettings.txt`** file is a text file that can be edited with Windows Notepad, Windows WordPad, or other text editor. Refer to **Figure 2** and **Table 4** in the following steps for information on the edits that are required.

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Steps	Description
2.	An excerpt of the **`46xxsettings.txt`** file with the SIP specific parameters is shown in **Figure 2**. A description of the important parameters is in **Table 4**. Edit these parameters to configure the SIP settings for the Avaya 4600 Series IP Telephones. **NOTE**: Lines that start with "**##**" are comments.

```
###################### SIP-Specific Settings ######################
## Use the following setting to configure SIP-specific parameters
##
## Examples:
## SET CALLFWDSTAT  "3"
## SET CALLFWDDELAY "5"
## SET CALLFWDADDR  "cover@avaya.com"
## SET COVERAGEADDR "cover@avaya.com"
## SET DIALPLAN     "[23]xxxx|91xxxxxxxxx|9[2-9]xxxxxxxxx"
## SET DIALWAIT     "5"
## SET MUSICSRVR    ""
## SET MWISRVR      "192.168.0.7"
## SET PHNNUMOFSA   "3"
## SET REGISTERWAIT "3600"
## SET SIPDOMAIN    "avaya.com"
## SET SIPPROXYSRVR "192.168.0.8"
## SET SIPPORT      "5060"
## SET SIPREGISTRAR "192.168.0.9"
## SET SPEAKERSTAT  "2"
## SET RTCPMON      "192.168.0.10"
## SET RTCPMONPORT  "5005"

SET DIALPLAN     "5xxxx"
## SET SIPDOMAIN    "sipsp.com"
SET SIPPROXYSRVR "10.2.2.50"
SET SIPREGISTRAR "10.2.2.50"
SET CALLFWDSTAT  "7"
SET CALLFWDADDR  "50020@avaya.com"
```

Figure 2: Excerpt from Sample Avaya 4600 Series IP Telephone 46xxsettings.txt File

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Steps	Description
3.	**Table 4** shows the SIP specific parameters that can be configured. The parameters that are critical to configure are **DIALPLAN**, **SIPDOMAIN**, **SIPROXYSRVR**, and **SIPREGISTRAR**. In addition, for testing local call forwarding, also configure **CALLFWDSTAT** and **CALLFWDADDR**. ▪ **CALLFWDSTAT**: This parameter defines which call forwarding buttons are configured on the phone. In this example, all three buttons (unconditional call forward [**Call Forward**], call forward on busy [**CFwrd-Busy**], and call forward on no answer [**CFrwd-DA**] are configured on the phone. On the phones, except the 4602SW, the buttons **Call Forward**, **CFwrd-Busy**, and **CFrwd-DA** will appear. ▪ **CALLFWDADDR**: This parameter defines the address to which calls will be forwarded when one of the call forwarding features are invoked. **NOTE**: The user can change this value using the interface on the phone. ▪ **DIALPLAN**: This parameter defines the dial plan used by the phone. In this example, "**5xxxx**" defines that extensions will start with the number 5 and will have a length of 5 digits. The server will automatically dial the number once the 5th digit is entered. ▪ ***SIPDOMAIN**: This parameter should be commented out (place "##" at the beginning of the line*[3] ▪ **SIPROXYSRVR**: This parameter is for the IP address of the proxy. ▪ **SIPREGISTRAR**: This parameter takes the same value as the proxy server (**SIPROXYSRVR**). **NOTE**: The **PHNUMOFSA** and **SPEAKERSTAT** default values do not apply to the 4602SW. The 4602SW telephone has a maximum of two (2) line appearances and the speakerphone works as a monitor or one-way speaker. The **CALLFWDSTAT** and **CALLFWDADDR** values also do not apply to the 4602SW.

SIP Setting	Parameter Name	Example Value
Call Forwarding Status	CALLFWDSTAT	7
Call Forwarding Delay	CALLFWDDELAY	5 (Default)
Call Forwarding Address	CALLFWDADDR	50020@sipsp.com
Forwarding Coverage Address	COVERAGEADDR	NULL [""] (Default)
Dial Plan	DIALPLAN	"5xxxx"
Dial Wait	DIALWAIT	5 (Default)
Music On Hold Server	MUSICSRVR	NULL [""] (Default)
Message Waiting Server	MWISRVR	NULL [""] (Default)
No of Line Appearances	PHNNUMOFSA	3 (Default)
Registration Expiration Timer	REGISTERWAIT	3600 (Default)
SIP Domain	*SIPDOMAIN*	*Comment out this parameter*
SIP Proxy Server	SIPPROXYSRVR	"10.2.2.50"
SIP Proxy Server Port	SIPPORT	5060 (Default)
SIP Registrar	SIPREGISTRAR	"10.2.2.50"
Speakerphone Status	SPEAKERSTAT	2 (Default)
RTCP Monitor	RTCPMON	NULL [""] (Default)
RTCP Monitor Port	RTCPMON	5005 (Default)

Table 4: Avaya 4600 Series IP Telephones SIP Parameters

[3] Normally, it is recommended to define the **SIPDOMAIN** parameter with a domain name, for example, "sipsp.com". However, testing showed that call attempts failed when the parameter was defined. When the **SIPDOMAIN** parameter was not defined, call attempts succeeded. No problems were observed with any of the features when this parameter was not defined.

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Steps	Description
4.	Connect the Avaya 4600 Series IP Telephone and reboot the phone. If DHCP, TFTP, and HTTP/HTTPS support have been properly configured, the phone will download the software and configuration files and prompt the user to for the extension and password: **Ext.=** **#=OK New=** Enter the extension followed by the **#** key. For example, **50000#**. **Password=** **#=OK** Enter the password followed by the **#** key. For example, **123456#**. On successful registration, the telephone's display will display something similar to the following: **9:50pm 4/2/06** **50000** The first line displays the date and the second line displays the extension number.
5.	Repeat step #4 for each of the Avaya 4600 Series IP Telephones.

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6. Configure the Avaya one-X™ Desktop Edition Software

The Avaya one-X™ Desktop Edition R2.1 software is available on the Avaya Support Center site (http://www.avaya.com/support). The installation and usage instructions for Avaya one-X™ Desktop Edition are documented in Reference [2]. The Avaya one-X™ Desktop Edition can be configured manually using the graphical user interface. As an option, there are some configurations items that can be configured via the **`46xxsettings.txt`** file. The steps for this configuration are documented in Reference [3] and further described in these Application Notes.

The configuration of Avaya one-X™ Desktop Edition via the **`46xxsettings.txt`** file requires the use of a DHCP server and an HTTP server. Avaya one-X™ Desktop Edition uses the same DHCP and HTTP mechanism used by the Avaya 4600 Series IP Telephones (see **Section 5**). These Application Notes will document only the steps to configure the DHCP server to support Avaya one-X™ Desktop Edition and the modifications to the **`46xxsettings.txt`** file in support of Avaya one-X™ Desktop Edition. **NOTE**: Using a DHCP and HTTP server to configure Avaya one-X™ Desktop Edition is optional. The information can be entered manually using the Avaya one-X™ Desktop Edition configuration wizard.

6.1. WebLM License Server

Avaya one-X™ Desktop Edition must connect to a WebLM license server and acquire a license from the license server. Without this license, Avaya one-X™ Desktop Edition will not support the full set of SIP features. The IP address of the license server is one of the options that can be gathered from the **`46xxsettings.txt`** file. The license server IP address can also be entered manually.

The WebLM server software can be installed on several operating systems (including Microsoft Windows and Linux). The WebLM software for these operating systems is available on the Avaya Support Center site (http://www.avaya.com/support). The installation of the WebLM software is described in Reference [3] but is not covered in these Application Notes.

WebLM is also installed as part of Avaya SIP Enablement Services. As these Application Notes describe testing the Avaya phones on a third party server, it is not expected that an SES would be part of the configuration. However, for simplicity, Avaya one-X™ Desktop Edition acquires its license by connecting to the WebLM server installed on an SES server (see **Figure 1**).

6.2. Configure the DHCP Server for Avaya one-X™ Desktop Edition (OPTIONAL)

The DHCP server that supports the Avaya 4600 Series IP Telephones can also be used to support Avaya one-X™ Desktop Edition as documented in Reference [1]. The DHCP server used in this testing is provided with Microsoft Windows 2000 Server. The examples shown in these Applications Notes are specific to this DHCP server implementation but also apply to other DHCP servers.

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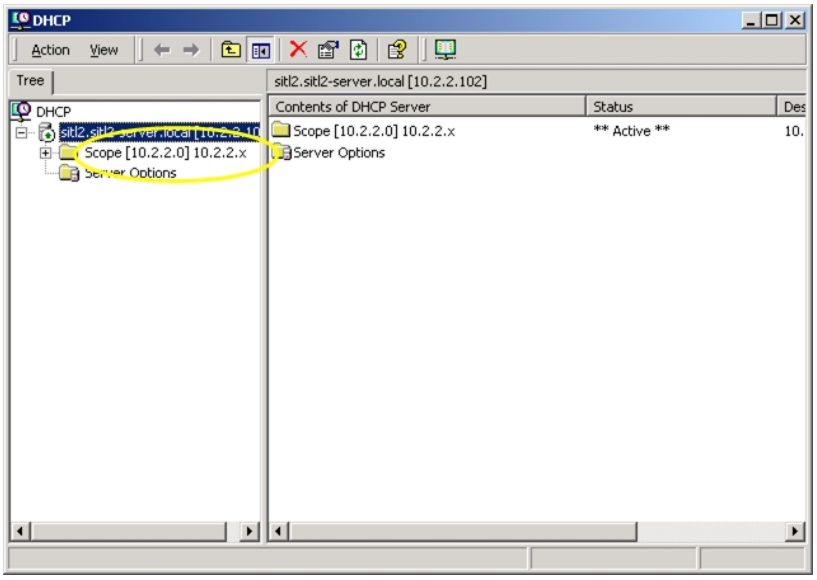
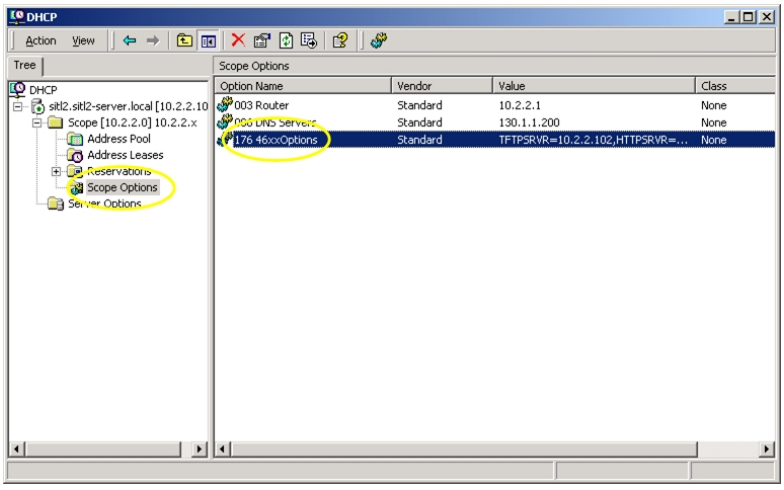
Steps	Description
1.	Edit the DHCP option configured for the Avaya 4600 Series IP Telephones. Start the DHCP server client from the Windows Control Panel (**Start → Settings → Control Panel**). Click on **Administrative Tools**. Click on **DHCP**. 
2.	Select **Scope Options** on the Scope created for the Avaya 4600 Series IP Telephones. In this example, select the scope for the *10.2.2.x* subnet. Select the scope option that was created for the Avaya 4600 Series IP Telephones on the right hand side of the DHCP window. In this example, select the scope option *176 46xxOptions*. From the menu, select **Action → Properties**. 

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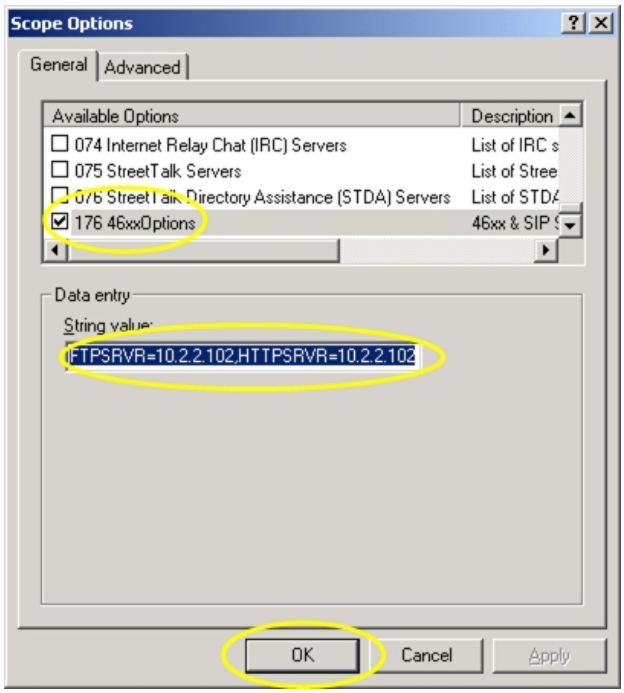
Steps	Description
3.	The *Scope Options* dialogue is displayed. The value for the *176 46xxOptions* option needs to be modified. The default string used by Avaya one-X™ Desktop Edition is in the form of: **`HTTPSRVR=nnn.nnn.nnn.nnn`** **`HTTPSRVR`** is the IP address of the server that holds the configuration script for Avaya one-X™ Desktop Edition. For this example, the option is set to the following. The **`TFTPSRV`** parameter is also set for the Avaya 4600 Series IP Telephones: **`TFTPSRVR=10.2.2.102,HTTPSRVR=10.2.2.102`** Click the **OK** button to save the change. 
4.	Refresh the DHCP server to use the updated scope options. To refresh the options, select the scope option (*176 46xxOptions*). Then from the menu, select **Action → Refresh**.

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6.3. Configure 46xxsettings.txt for Avaya one-X™ Desktop Edition (OPTIONAL)

Assuming that the DHCP server has been configured to support Avaya one-X™ Desktop Edition (see **Section 6.2**), the **`46xxsettings.txt`** configuration file must be modified to provide the Avaya one-X™ Desktop Edition configuration information.

Steps	Description
1.	Locate the **`46xxsettings.txt`** file to administer the SIP specific settings for Avaya one-X™ Desktop Edition. This file is stored on the PC that provides HTTP services to Avaya one-X™ Desktop Edition. The location of this file depends on the software that is used to provide these services. For these Application Notes, the file is stored on the **`Inetpub\wwwrooot`** folder on the Windows server PC. This is a text file and can be edited with Windows Notepad, Windows WordPad, or other text editor. Refer to **Figure 3** and **Table 5** in the following steps for information on the edits that are required.

2.	An excerpt of the **`46xxsettings.txt`** file with the SIP specific parameters is shown in **Figure 3**. A description of the important parameters is in **Table 5**. Edit these parameters to configure the SIP settings for Avaya one-X™ Desktop Edition.
	
	NOTE: Lines that start with "**##**" are comments.
	
	`############################################################`
	`## Avaya one-X™ Desktop Edition Variables`
	`############################################################`
	`##`
	`## Examples:`
	`## SET SIPPROXYSRVR 192.168.0.10`
	`## SET WEBLMSRVR 192.168.0.11`
	`## SET SP_DIRSRVR ldap-east.post.avaya.com`
	`## SET SP_DIRSRVRPORT 389`
	`## SET SP_DIRTOPDN ou=People,o=avaya.com`
	`## SET SP_AC 212`
	`## SET LOCAL_CALL_PREFIX AC`
	`## SET SIPPROXYSRVR`
	`## SET PHNCC`
	`## SET PHNDPLENGTH`
	`## SET PHNIC`
	`## SET PHNLD`
	`## SET PHNLDLENGTH`
	`## SET PHNOL`
	
	`SET SIPPROXYSRVR 10.2.2.50`
	`SET WEBLMSRVR 10.2.2.70`
	
	Figure 3: Excerpt from Sample Avaya one-X™ Desktop Edition 46xxsettings.txt File

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Steps	Description
3.	**Table 5** shows the SIP specific parameters that can be configured. The parameters that are critical to configure are **SIPPROXYSRVR** and **WEBLMSRVR**. ▪ **SIPPROXYSRVR**: Enter the IP address of the Cisco proxy server. ▪ **WEBLMSRVR**: Enter the address of the WebLM server.

SIP Setting	Parameter Name	Example Value
SIP Proxy Server	SIPPROXYSRVR	"10.2.2.50"
WebLM Server	WEBLMSRVR	"10.2.2.70"
LDAP Directory Server	SP_DIRSRVR	5 (Default)
LDAP Directory Server Port	SP_DIRSRVRPORT	NULL [""] (Default)
LDAP Directory Topmost Distinguished Name	SP_DIRTOPDN	NULL [""] (Default)
Area Code	SP_AC	NULL [""] (Default)
Local Call Prefix	LOCAL_CALL_PREFIX	"DIAL_AS_IS" (Default)
Country Code	PHNCC	1 (Default)
Dial Plan Length	PHNDPLENGTH	NULL [""] (Default)
International Access Code	PHNIC	011 (Default)
Long Distance Access Code	PHNLD	1 (Default)
National Telephone Length	PHNLDLENGTH	10 (Default)
Outside Line Access Code	PHNOL	9 (Default)

Table 5: Avaya 4600 Series IP Telephones SIP Parameters

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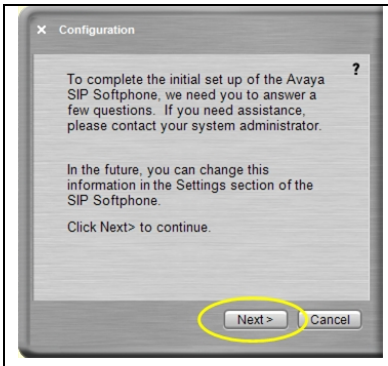
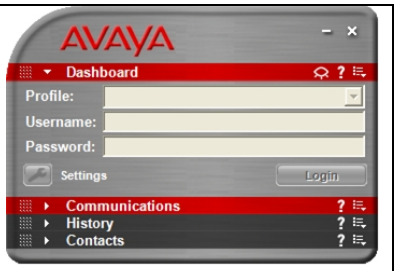
6.4. Configure Avaya one-X™ Desktop Edition

After Avaya one-X™ Desktop Edition is installed, it must be configured.

Steps	Description
1.	Start Avaya one-X™ Desktop Edition via the menu **Start → Programs → Avaya SIP Softphone → Avaya SIP Softphone**.[4] The first time the program is executed, the Configuration Wizard will be displayed. Edit the **`46xxsettings.txt`** file to administer the Avaya one-X™ Desktop Edition specific settings. The main *Avaya one-X™ Desktop Edition* dialogue and the *Configuration* dialogue are displayed. Click the **Next** button to continue.  

[4] While the name of the product has changed to "Avaya one-X™ Desktop Edition", the software used for testing still refers to "Avaya SIP Softphone". This includes the installation folder and Windows program group.

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Steps	Description
2.	The *Account* dialogue is displayed. Enter the name for the Avaya one-X™ Desktop Edition station. This name will be used for caller identification. Enter the user name that will be used to authenticate with the SIP registrar. **NOTE**: It is recommended to use a domain name instead of an IP address for the user name. For this example, the name is **50060@sipsp.com** instead of **50060@10.2.2.50**. Choose the appropriate option for the password. Click the **Next** button to continue.
3.	The *SIP Server/License Server* dialogue is displayed. If the DHCP server was configured properly in **Section 6.2** and the **`46xxsettings.txt`** file has the proper information as described in **Section 6.3**, Avaya one-X™ Desktop Edition will get the IP address of the proxy server and the license server. Click the **Next** button to continue.

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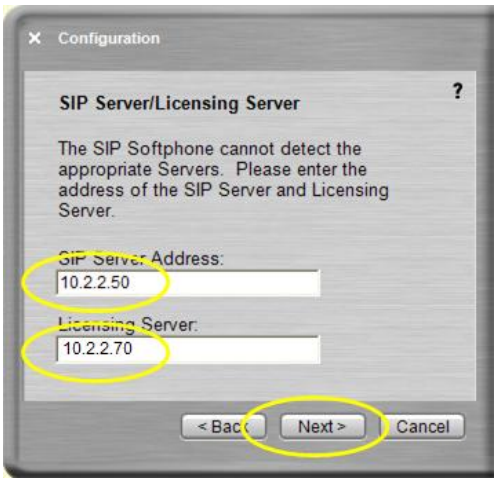
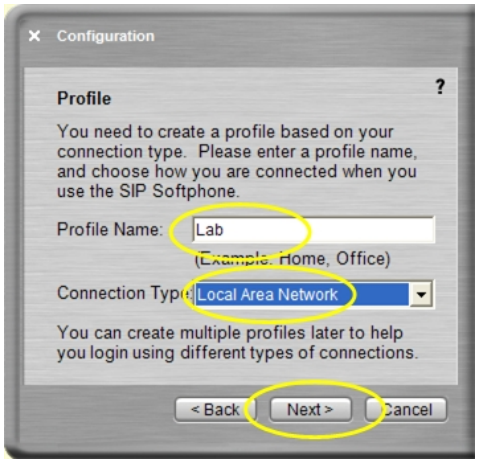
Steps	Description
4.	An updated *SIP Server/License Server* dialogue is displayed. If the information was not gathered from the DHCP & HTTP servers, the fields would be blank and have to be entered them manually. Click the **Next** button to continue. 
5.	The *Profile* dialogue is displayed. For this example, **Lab** was entered for the *Profile Nam*e. Any name can be used. Select **Local Area Network** for the *Connection Type*. The other options are **Cable, xDSL, or IDSN** and **Modem (28800 bps or faster)**. Click the **Next** button to continue. 

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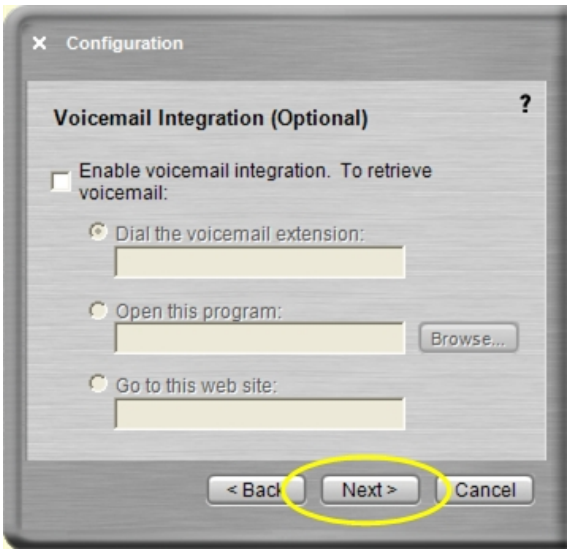
Steps	Description
6.	The *Dialing Rules* dialogue is displayed. If DHCP and HTTP are configured and appropriate values were entered in the **`46xxsettings.txt`** file, those values are populated in this dialogue. Click the **Next** button to continue.
7.	The *Voicemail Integration* dialogue is displayed. Voice mail is not covered in these Application Notes. Click the **Next** button to continue. 

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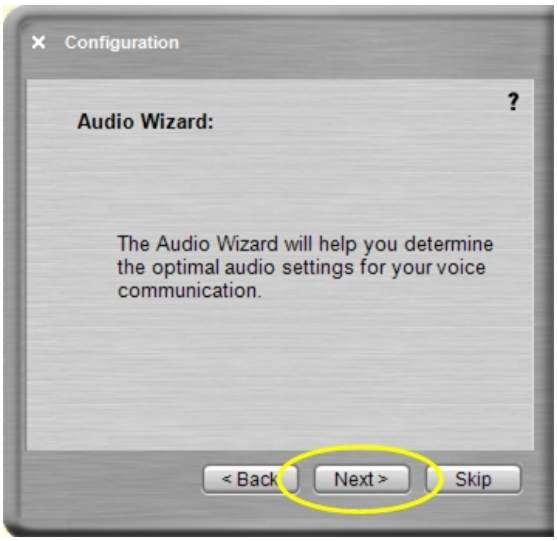
Steps	Description
8.	The *Audio Wizard* dialogue is displayed. Click the **Next** button to continue. 

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Steps	Description
9.	The *Audio Wizard: Select Sound Device* dialogue is displayed. The recommendation is to use a USB headset with Avaya one-X™ Desktop Edition. A list of recommended headsets is available on the Avaya Support Center (http://www.avaya.com/support). Depending on how many sound devices are installed or connected to the PC (e.g., built-in sound device, USB headset), there may be one or more audio output (speaker) and audio input (microphone) devices available. Select a sound device from the pull-down list to be used for the audio output. Select a microphone from the pull-down list to be used for the audio input. Click the **Next** button to continue.

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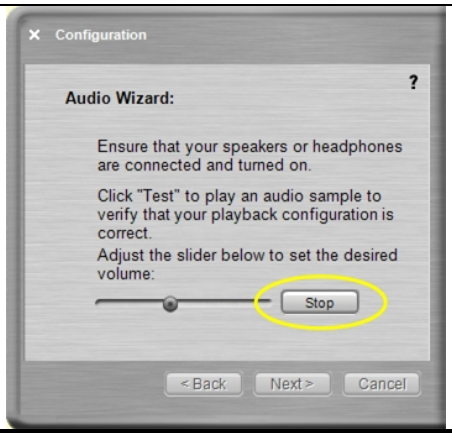
Steps	Description
10.	The *Audio Wizard: Test Speaker* dialogue is displayed. Click the **Test** button and adjust the volume with the slider. The **Test** button changes to a **Stop** button. Click the **Stop** button once the proper volume for playback is achieved. Click the **Next** button to continue. 
11.	The *Audio Wizard: Tune Microphone* dialogue is displayed. Talk into the microphone and adjust the volume with the slider. Click the **Next** button to continue.

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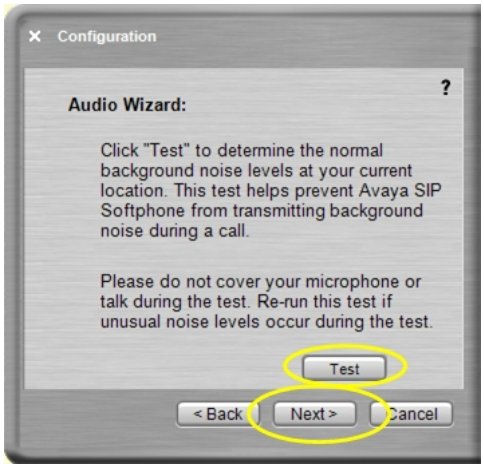
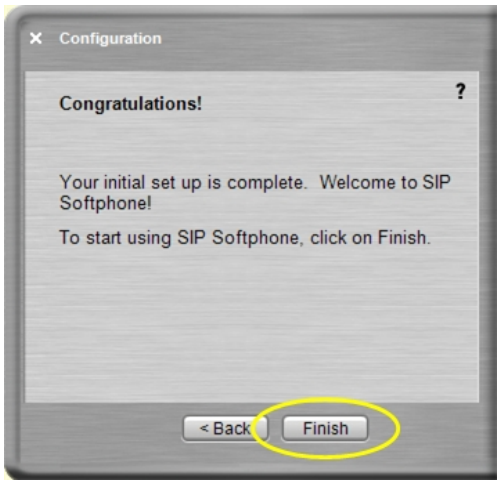
Steps	Description
12.	The *Audio Wizard: Test Background Noise* dialogue is displayed. Click on the **Test** button. Do not talk during this test. Click the **Next** button to continue. 
13.	The *Congratulations* dialogue is displayed. The configuration is complete. Click the **Finish** button to continue. 

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Steps	Description
14.	The *Avaya one-X™ Desktop Edition* dialogue is displayed again, with its configuration information populated. Before Avaya one-X™ Desktop Edition can be used with the Cisco proxy server, it must be optioned to use UDP to communicate with the server. Click the **Settings** button.

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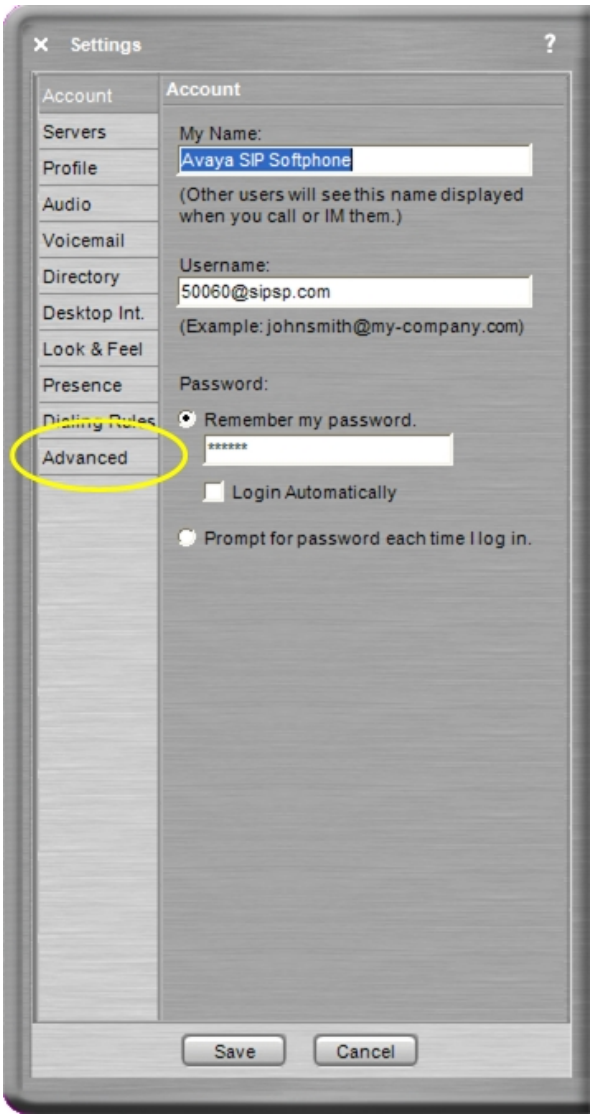
Steps	Description
15.	The *Settings: Account* dialogue is displayed. Click on **Advanced**. 

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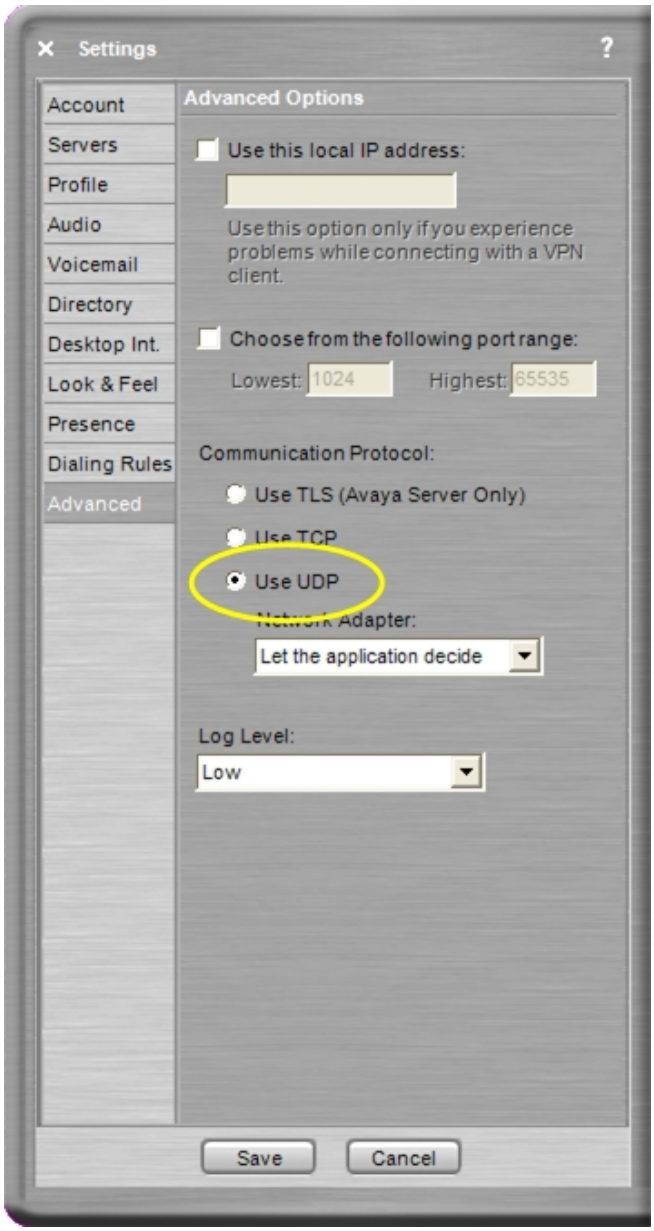
Steps	Description
16.	The *Settings: Advanced Options* dialogue is displayed. Select **Use UDP** for the **Communications Protocol**. 

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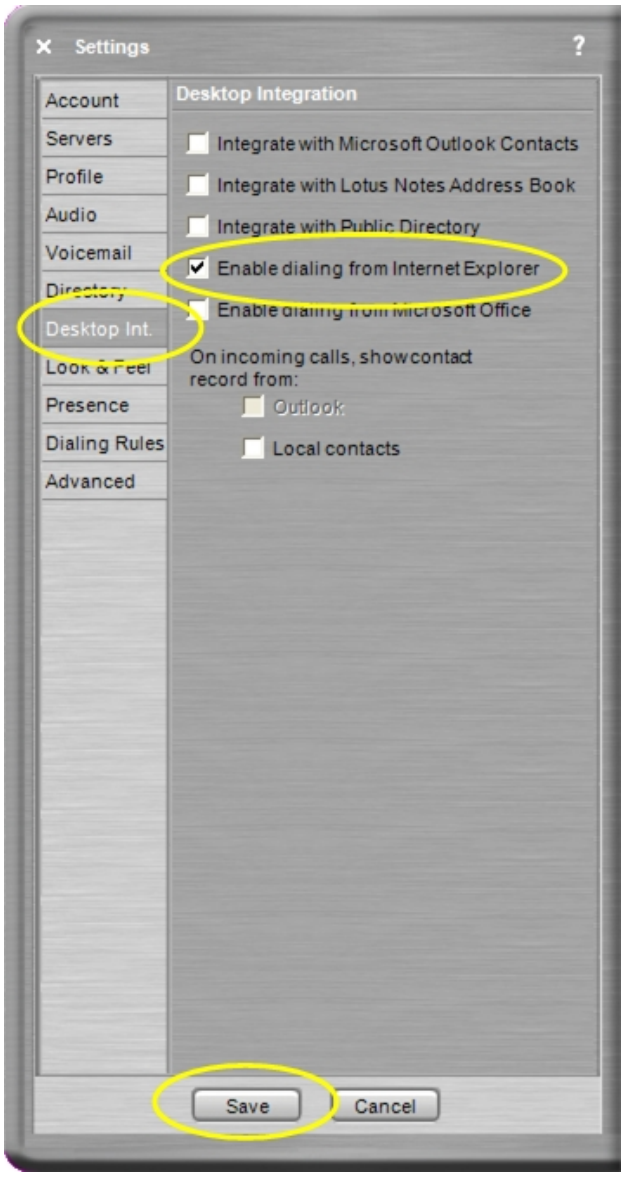
Steps	Description
17.	Avaya one-X™ Desktop Edition must be configured to support Click to Dial using Microsoft Internet Explorer. Click **Desktop Int.** from the *Settings* dialogue. Enable the option **Enable dialing from Internet Explorer**. Click the **Save** button to continue. 

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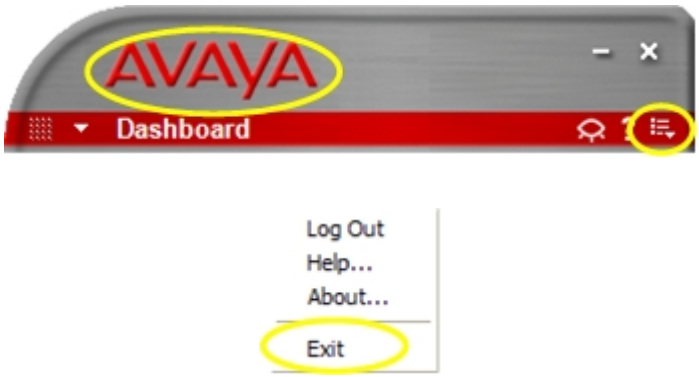
Steps	Description
18.	A warning message is displayed indicating that Avaya one-X™ Desktop Edition must be re-started before the changes can take place. Click the **OK** button.
19.	Click on the Avaya logo on the *Avaya one-X™ Desktop Edition* dialogue and a pop-up menu is displayed. Alternatively, right-click on the far right of the Dashboard to bring up the pop-up menu. Click **Exit** to exit Avaya one-X™ Desktop Edition. 

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7. Verification Steps

All features shown in **Table 3** that have a "**Yes**" in the *Supported* column were tested. Two problems were found.

1. When the **SIPDOMAIN** parameter was defined in the **46xxsettings.txt** file, calls attempts from the Avaya 4600 Series IP Telephones failed. The Cisco server replies that the destination could not be found. However, calls to the Avaya 4600 Series IP Telephones succeeded. For the testing documented in these Application Notes, the **SIPDOMAIN** parameter was not defined (commented out). With this change, call attempts from the Avaya 4600 Series IP Telephones succeeded.
2. Avaya one-X™ Desktop Edition subscribes to a presence-list event that is supported by the Avaya SIP Enablement Services 3.1. Through this presence-list event, the Avaya client, Avaya one-X™ Desktop Edition, knows which other clients should be tracked and which other clients should be blocked from tracking its presence state. The Avaya client can be optioned to block one or more clients from tracking its presence state (**Settings → Presence → Presence Control**). The Cisco SIP proxy server does not support the presence-list event. Therefore, when the Avaya client was optioned to block others from tracking its presence, the other clients were still able to track its presence.

The following steps can be used to verify and/or troubleshoot installations in the field.

- Avaya 4600 Series IP Telephones
  1. If there are problems with calls attempts, verify that the **SIPDOMAIN** parameter is **NOT** defined in the **46xxsettings.txt** configuration file.
  2. If there are problems with receiving calls, verify that call forwarding has not been locally configured and enabled. This may impact receiving calls depending on which forwarding feature is enabled. This configuration is done in the **46xxsettings.txt** configuration file (**CALLFWDADDR** and **COVERAGEADDR** parameters). However, the call forwarding features can be disabled at the telephone.
- Avaya one-X™ Desktop Edition
  1. Verify that the Username field for Avaya one-X™ Desktop Edition is in the format of **userID@domain-name** instead of **userID@IP-address** (e.g., john.doe@company.org instead of john.doe@123.22.33.444). If an IP address is used, this may affect the ability to complete transfers.
  2. Verify that Avaya one-X™ Desktop Edition has been configured for UDP (**Settings → Advanced → Communications Protocol**). Otherwise, it will not register with the server.
  3. Verify that Avaya one-X™ Desktop Edition is able to connect to a WebLM license server and acquire a license. Otherwise, only basic telephony (make & receive calls) will work. From the local PC, open a Command Prompt window and ping the license server. From the license server, verify that there are available SIP Softphone licenses installed.

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4. Verify that the Dialing Rules are properly configured (**Settings → Dialing Rules**). When improperly configured, external calls may be dialed incorrectly. From the same dialogue (**Settings → Dialing Rules**), enable the **Display confirmation window before dialing a number** to view the number that will be dialed before it is actually dialed.

- Cisco SIP Proxy Server

1. If there are problems with receiving calls, verify that call forwarding has not been configured and enabled on the server. This may impact receiving calls depending on which forwarding feature is configured. From the Cisco SIP Proxy click **Subscribers** and double-click on the appropriate subscriber to see if forwarding is configured.

8. Conclusion

These Application Notes have described the administration steps required to use Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition with the Cisco SIP Proxy Server. Both standard and supplementary features were tested for the Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition and presence tracking & instant messaging were tested for Avaya one-X™ Desktop Edition.

9. Additional References

The following are additional references.

[1] *4600 Series IP Telephone R2.3 LAN Administrator Guide*, Issue 2.3, Doc ID 555-233-507, April 2006, available at http://www.avaya.com/support.

[2] *Avaya one-X™ Desktop Edition R2.1 Getting Started Guide, February 2006. Avaya one-X™ Desktop Edition Overview*, April 2006, available at http://www.avaya.com/support.

[3] *Avaya one-X™ Desktop Edition Administration*, April 2006, available at http://www.avaya.com/support.

[4] *Session Initiation Protocol Service Examples - draft-ietf-sipping-service-examples-10*, SIPPING Working Group, Internet-Draft, Mach 5, 2006 (expires September 6, 2006), available at http://tools.ietf.org/wg/sipping/draft-ietf-sipping-service-examples/draft-ietf-sipping-service-examples-10.txt.

[5] Cisco SIP Proxy Server Version 2.2 Administrator Guide, Version 2.2, available at http://www.cisco.com/en/US/support/.

[6] Cisco SIP Proxy Server Version 2.2 Installation Guide, Version 2.2, available at http://www.cisco.com/en/US/support/.

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